

FREQUENCY SYNCHRONIZER

BACKGROUND OF THE INVENTION

The present invention generally relates to frequency synchronizers for digital communications systems, and more particularly, to frequency synchronizers that are based on the maximum likelihood criterion from estimation theory and that can achieve both frequency acquisition and frequency tracking without requiring knowledge at the receiver of the carrier's phase angle, baud timing, or a preamble consisting of known signal symbols.

All digital communications systems operate on the basis of a finite number of possible waveforms available for transmission during any particular signaling interval. The digital receivers must process versions of these transmitted waveforms which are corrupted by noise, channel fading, multipath, distortion, unintentional interference, and jamming, for example. The receiver's task is to determine which was the transmitted waveform for a particular signaling interval. Acceptable performance requires that this determination be achieved with high probability of correctness.

The degree of success potentially achievable by a digital communications system depends on the accuracy of reference signals at the receiver in their representation of the possible transmitted waveforms, as they would appear at the receiver, including the effects of noise, etc. To a large degree, achieving good reference signals is synonymous with having the receiver be synchronized with the transmitted waveforms arriving over a transmission channel.

Synchronization in a particular case may involve several parameters. For example, the receiver's reference signals should be based on the correct carrier frequency, which may be unknown due to oscillator drifts or doppler shifts. For best performance, the receiver should have a timing reference to know the beginning of each signaling waveform. For Time-Division-Multiplex (TDM) and/or Time-Division-Multiple Access (TDMA) systems, the level of timing information must be extended to knowing the beginning of groups of time slots (frame synchronization). In order to have the performance improvement potentially available from

coherent detection; the receiver requires accurate knowledge of the carrier's phase angle (phase
synchronization). For spread-spectrum systems, synchronization to a hopping-frequency pattern
and/or a spread-spectrum code sequence is required. For all levels of synchronization, typically
the receiver's synchronizers are required to provide good estimates of the unknowns (frequency,
phase, timing, etc.) during an initial start-up period (so-called "acquisition") and to continue to
provide good estimates as the system proceeds to operate (so-called "tracking").

Two general approaches have been useful for designing synchronizers for digital
receivers. Many existing synchronizers are based on good engineering reasoning as to what can
work (so-called "ad hoc" procedures), as opposed to being mathematically derived based on
various math models and theoretical reasoning. Synchronizers of the latter type typically are
derived, and implemented, by using the tools of Estimation Theory based upon the Maximum-
Likelihood criterion of goodness (maximization of appropriate conditional probability density
functions).

The choice and/or design of synchronizers for a particular system greatly depends on the
digital modulation technique to be employed and the channel over which the communication is
to take place. Practical solutions have long existed for conventional digital modulation
techniques such as Phase Shift Key (PSK), Frequency Shift Key (FSK), Amplitude Shift Key
(ASK), and Quadrature Amplitude Modulation (QAM), particularly for the case of a Gaussian
noise channel. To meet requirements of transmitting data at high rates, with high accuracy, and
with minimal bandwidth usage, newer digital signaling techniques have been found. These
include Trellis-Coded Modulation (TCM) and Continuous-Phase Modulation (CPM). Generally,
the synchronizers for the older, conventional digital receivers are inadequate for the newer
techniques. In fact, there are many theoretical versions (special cases) of the energy-efficient and
bandwidth-efficient CPM which would be preferred choices for applications but for the lack of
good, achievable synchronizers. Since general solutions are unknown, each category of CPM
requires finding specialized solutions for the synchronizer designs required for system operation.

An existing system for which better synchronizers are desirable is a special version of CPM known as dual-h, 4-ary, full-response. The meaning of these terms follows from the mathematical model of a signal having the specified CPM waveform, $s(t)$, given below.

$$s(t) = \sqrt{\frac{2E_s}{T}} e^{j\Psi(t, \underline{\alpha})}$$

with E_s representing the waveform's energy over its interval T , and $\Psi(t, \underline{\alpha})$ is the phase function.

The function $\Psi(t, \underline{\alpha})$ depends on the data sequence $\underline{\alpha} = (\dots \alpha_{i-1}, \alpha_i, \alpha_{i+1}, \dots)$ where each of the data symbols is randomly and independently selected from the four possibilities $(\pm 1, \pm 3)$, hence "4-ary." Also, $\Psi(t, \underline{\alpha})$ depends on two constants h_0 and h_1 , called "modulation indexes," and a function $q(t)$, called the "phase response function," as follows.

$$\begin{aligned} \Psi(t, \underline{\alpha}) = & 2\pi h_0 \sum_i \alpha_{2i} q(t - 2iT) \\ & + 2\pi h_1 \sum_i \alpha_{2i+1} q(t - 2i - T) \end{aligned}$$

For "full response" CPM, the function $q(t)$, is as shown in Figure 1.

A method for synchronizing for the above CPM waveform case requires transmitting a preamble at the beginning of a message. The preamble is a sequence of non-data symbols, known to the receiver. This preamble sequence is transmitted by means of a modulation technique, less complex than the CPM used for data, called Minimum Shift Key (MSK). However, this method of synchronization has certain undesirable characteristics. For example, the requirement that a start-up interval be set aside for a known preamble means a reduction in information rate. A serious problem arises if the receiver is unable to detect the preamble, thus leading to the loss of the follow-on message. Another potential problem of great concern when operating in the presence of an adversary is that the use of a different modulation for a preamble from that for data offers the adversary significant information useful for a jamming attack.

Therefore, a need exists for synchronization processors/circuits that operate without a preamble and without changing modulation methods within a transmission.

SUMMARY OF THE INVENTION

2 The present invention provides a frequency synchronizer that does not require a
transmitted known-sequence preamble prior to the transmission of information symbols as a
4 message. Nor does the synchronizer require that the receiver be synchronized in time or carrier
phase prior to its operation to obtain information allowing the receiver to become synchronized
6 in frequency, so-called frequency acquisition. In addition to frequency acquisition, the frequency
synchronizer provides the receiver with continual updates on reference-frequency changes during
8 the time of message transmission, so-called frequency tracking.

 The frequency synchronizer is based on the maximum likelihood criterion from
10 estimation theory and that can achieve both frequency acquisition and frequency tracking
without requiring knowledge at the receiver of the carrier's phase angle, baud timing, or a
12 preamble consisting of known signal symbols. The synchronizer is part of a system that includes
a processor for executing the following sequence of operations: a) initializing an estimated
14 frequency correction factor, as for example, at zero; b) determining a corrected frequency offset
value from a first product of a sample signal and the estimated frequency correction factor; c)
16 filtering a first sample of the corrected frequency offset value to obtain a filtered corrected
frequency offset value; d) imparting a delay to a second sample of the corrected frequency offset
18 value to obtain a delayed corrected frequency offset value; e) determining a conjugate product
value from a second product of the filtered corrected frequency offset value and a conjugate of
20 the filtered corrected frequency offset value; f) determining a delay conjugate value from a third
product of the delayed corrected frequency offset value and the conjugate product value; g)
22 determining an error signal from the delay conjugate value; h) determining a frequency offset
value from the error signal; and i) determining an updated value of the estimated frequency
24 correction factor from the frequency offset value.

The frequency synchronizer is applicable for use in conjunction with receivers that detect communications signals that belong to a class of digital modulation waveforms known as dual-h, 4-ary, full-response, CPM.

Since the frequency synchronizer of the invention operates without a transmitted preamble, the energy efficiency and the bandwidth efficiency of the communications system are greater than for systems that require preambles. An important advantage of the invention is that it obviates the need for different modulation methods for signals that employ preambles.

Other advantages of the invention will become apparent upon review of the following specification, including the claims, and the accompanying figures.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 illustrates the function $q(t)$ for a "full response" Continuous-Phase Modulation (CPM) signal.

Figure 2 is a block diagram representing a method for implementing a frequency synthesizer that embodies the various features of the present invention.

Figure 3 shows a digital data processor for implementing the method depicted in Figure 2 and for providing a corrected frequency offset value to a digital receiver.

Throughout the several views, like elements are referenced using like references.

DESCRIPTION OF THE PREFERRED EMBODIMENT

The present invention is directed to a frequency synchronizer for synchronizing to a Continuous-Phase Modulation (CPM) signal that does not require phase or timing information from a detected signal prior to processing for frequency estimation. Nor does the invention require that there be a preamble of known symbols at the beginning of a transmitted sequence. Referring to Figure 2, there is shown a block diagram of a method for implementing a frequency synthesizer 10 that embodies several features of the present invention. A received signal 12 consisting of a CPM signal $s(t)$ of the dual-h, 4-ary, full-response type and an additive white

Gaussian noise component $w(t)$ may be represented as a function of time t by the equation
 $r(t) = s(t) + w(t)$. Signal 12 is transformed by an anti-aliasing filter 14 into a time dependent
signal 16 represented by $x(t)$. The anti-aliasing filter 14 essentially removes from signal 12 those
frequency components not essential for maintaining information contained in CPM signal, $s(t)$,
thereby reducing the inherent distortion produced by sampling time-limited analog waveforms.
Signal 16 is sampled by sampler 15 at intervals spaced T_s seconds apart; i.e., at, a sampling rate
of $1/T_s$ samples per second. The sampler 15 is represented as a switch that opens and closes at a
frequency $1/T_s$. The signal 18 emitted by sampler 15 is a set of discrete time samples represented
by $x(kT_s)$, where k is an integer index so that $x(kT_s)$ is the k^{th} sample of the set $x(kT_s)$.

Signal 22 is a corrected frequency offset value that is represented by $y(kT_s)$, also a set of
discrete time samples, produced by multiplying signal 18 [$x(kT_s)$] by signal 24 at multiplication
node 20, where signal 24 represents an estimated frequency correction factor. At the
initialization of the operation of frequency synchronizer 10, signal 24 may be provided with an
initial value of zero. Signal 24 is a discrete time sequence represented by $e^{-j2\pi k\hat{\nu}T_s}$. The resultant
product from node 20 is signal 22, and represents a corrected version of signal 18 [$x(kT_s)$], with
the correction taking the form of subtracting from signal 18 an estimate $\hat{\nu}$ produced by
synchronizer 10 of the unknown frequency offset ν from the actual carrier frequency of CPM
signal $s(t)$. Next, signal 22, i.e., the sequence $y(kT_s)$, is filtered by a digital filter 26 having
impulse response $h(kT_s)$. The impulse response $h(kT_s)$ is specified by the following equations:

$$\begin{aligned} h(kT_s) &= kF_2(kT_s), \text{ for } -N \leq k \leq -1 \\ &= kF_3(kT_s), \text{ for } 1 \leq k \leq N \\ &= 0, \text{ for } k = 0 \text{ and } |k| > N, \end{aligned}$$

where N represents the number of samples for each information symbol interval, and where F_2
and F_3 are functions defined as follows:

$$\begin{aligned}
F(kT, kT) = & \frac{1}{4} \left[1 + \frac{(k_2 - k_1)}{N} \right] \left[\cos \frac{3\pi h_0 (k_2 - k_1)}{N} + \cos \frac{\pi h_0 (k_2 - k_1)}{N} \right. \\
& + \left. \cos \frac{3\pi h_1 (k_2 - k_1)}{N} + \cos \frac{\pi h_1 (k_2 - k_1)}{N} \right] \\
& + \frac{1}{4\pi} \left\{ \left[\frac{h_1}{3(h_0^2 - h_1^2)} + \frac{3h_1}{h_0^2 - 9h_1^2} \right] \sin \frac{3\pi h_1 (k_2 - k_1)}{N} \right. \\
& + \left[\frac{h_1}{9h_0^2 - h_1^2} + \frac{h_1}{h_0^2 - h_1^2} \right] \sin \frac{\pi h_1 (k_2 - k_1)}{N} \\
& + \left[\frac{h_0}{9h_1^2 - h_0^2} + \frac{h_0}{h_1^2 - h_0^2} \right] \sin \frac{\pi h_0 (k_2 - k_1)}{N} \\
& + \left. \left[\frac{h_0}{3h_1^2 - h_0^2} + \frac{3h_0}{h_1^2 - 9h_0^2} \right] \sin \frac{3\pi h_0 (k_2 - k_1)}{N} \right\} \\
= & F_2[(k_2 - k_1)T_s]
\end{aligned}$$

and where:

$$\begin{aligned}
F(kT, kT) = & \frac{1}{4} \left[1 - \frac{(k_2 - k_1)}{N} \right] \left[\cos \frac{3\pi h_0 (k_2 - k_1)}{N} + \cos \frac{\pi h_0 (k_2 - k_1)}{N} \right. \\
& + \left. \cos \frac{3\pi h_1 (k_2 - k_1)}{N} + \cos \frac{\pi h_1 (k_2 - k_1)}{N} \right] \\
& + \frac{1}{4\pi} \left\{ \left[\frac{h_1}{3(h_1^2 - h_0^2)} + \frac{3h_1}{9h_1^2 - h_0^2} \right] \sin \frac{3\pi h_1 (k_2 - k_1)}{N} \right. \\
& + \left[\frac{h_1}{h_1^2 - 9h_0^2} + \frac{h_1}{9h_1^2 - h_0^2} \right] \sin \frac{\pi h_1 (k_2 - k_1)}{N} \\
& + \left[\frac{h_0}{h_0^2 - 9h_1^2} + \frac{h_0}{h_0^2 - h_1^2} \right] \sin \frac{\pi h_0 (k_2 - k_1)}{N} \\
& + \left. \left[\frac{h_0}{3h_0^2 - h_1^2} + \frac{3h_0}{9h_0^2 - h_1^2} \right] \sin \frac{3\pi h_0 (k_2 - k_1)}{N} \right\} \\
= & F_3[(k_2 - k_1)T_s]
\end{aligned}$$

Since filter 26 is non-causal, a time delay of D sampling intervals is incorporated into $h(kT_s)$ of filter 26. By way of example, D may be set equal to 2N, but other choices are possible as might be suitable for particular applications. The filter 26 is thus shown as $h[(k - D)T_s]$ so as to incorporate the necessary delay. The output of filter 26 is a signal 28 represented as a discrete

time sequence $w(kT_s)$, which is transformed at step 29 into signal 29, the conjugate sequence $w^*(kT_s)$. Signal 28 represents a filtered corrected frequency offset value. Signal 22 $[y(kT_s)]$ is also delayed D sampling intervals by the delay 32, which produces signal 33, represented as $y[(k-D)T_s]$ and described as a delayed corrected frequency offset value. The multiplier 34 forms a signal 36 which is a discrete time sequence and is the product of signals 30 and 33, i.e., $w^*(kT_s) y[(k-D)T_s]$. Signal 30 is described as a conjugate product value. Signal 36, a delay conjugate value, is provided as an input to error generator 38.

Error generator 38 transforms signal 36 into error signal 40 which is a discrete time sequence $e(nT)$, with n being the positive integer index allowing for the generation of updated error signals at intervals of T seconds. T is the time for each separate CPM waveform and is NT_s . The error signal 40 $[e(nT)]$ for a particular value of n is calculated by error generator 38 from the summing of N real numbers comprising signal 36, as follows:

$$e(nT) = \sum_{k=nN}^{(n+1)N-1} I_m \{y[(k-D)T_s] w^*(kT_s)\}.$$

The error signal 40 is used by loop filter 42 to produce signal 44, a frequency offset value, which is a sequence $\hat{v}(nT)$ of estimates of the unknown frequency offset V. The sequence $\hat{v}(nT)$ is produced iteratively by the discrete time loop filter 42 which is a first-order filter described by the difference equation

$$\hat{v}[(n+1)T] = \hat{v}(nT) + \gamma e(nT).$$

Thus, the next estimate $\hat{v}[(n+1)T]$ of the unknown frequency offset is the present estimate of $\hat{v}(nT)$ added to the error signal 40 $[e(nT)]$ that is weighted by a step-size parameter constant γ selected from computer simulations for a particular application. The choice of γ is based upon a need to have the sequence $\hat{v}(nT)$ converge toward v at a reasonable rate without having large overshoots.

Signal 44, represented as the sequence $\hat{v}(nT)$, is converted by a discrete time voltage controlled oscillator (VCO) 46 into signal 24, which was previously described above as the

sequence $e^{-j2\pi k\hat{\nu}T_s}$ for updating and correcting signal 18 $[x(kT_s)]$. The frequency synchronizer
10 is, therefore, a discrete time closed-loop processor, suitable for both frequency acquisition and
frequency tracking.

Referring to Figure 3, frequency synchronizer 10 may be implemented as a sequence of
executable operations in a discrete time digital data processor 50 which provides signal 22 (a
corrected frequency offset) to a digital receiver 52, which outputs a signal 54. Signal 54 is a data
sequence that represents an improved estimate of the data encoded in signal 12, or $s(t)$. The
operation of frequency synchronizer 10 may be repeated any integral number of times to provide
increasingly refined values for the corrected frequency offset value as represented by signal 22.

Obviously, many modifications and variations of the present invention are possible in
light of the above teachings. It is therefore to be understood that within the scope of the
appended claims, the invention may be practiced otherwise than as specifically described.